



DAG1000/2000 Series FXSFXO Voice Gateway

User Manual V2.0



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1. Equipment Introduction

1.1 Overview

Thanks for purchasing Dinstar DAG1000/2000 (Hereinafter referred to as the DAG) series FXS/FXO hybrid analog voice gateway. DAG1000/2000 series FXS/FXO hybrid analog gateway is access gateway based on IP network. It can provide low cost, simple operation VoIP solutions for small enterprise, the family office, remote office and branch enterprise. DAG connects to analog telephone, fax and traditional analog PBX with standard voice interfaces and provided high quality voice service. DAG series FXS/FXO hybrid VoIP access gateway has life line function. DAG series FXS/FXO hybrid gateway includes following model:

- DAG1000-4S4O
- DAG1000-8S8O

This manual mainly to DAG1000-4S4O as examples, introduce the function of devices and parameter configuration.

1.2 Equipment appearance



Figure 1-1DAG1000-4S4O



Figure 1-2 DAG1000-8S8O

1.3 Power supply

DAG1000-4S4O is Cassette equipment with placed on desk, and adopts AC 110-240V power supply, with the power adapter convert to 12V DC power.

Power parameters:

Input: 100-240VAC, 50-60Hz

Output: 12V DC

DAG2000-8S8O can be installed in the 19 inch frame, and adopts AC power supply.

Notes: Because power adapter interface is different in different country, please confirm the interface standard with us before shipment.

1.4 Network Applications

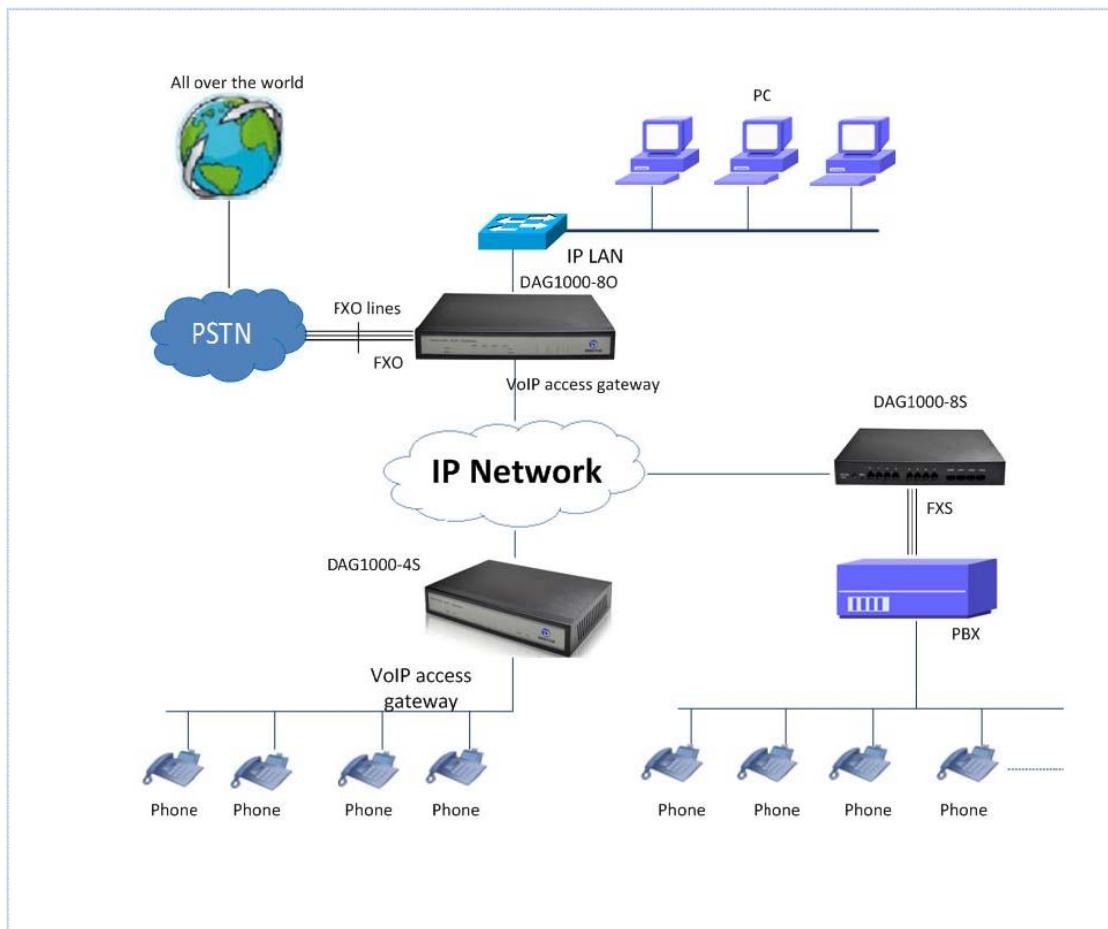


Figure 4-1: Network Applications

1.5 Functions and Features

1.5.1 Protocol standard supported

- SIP V2.0 (RFC 3261,3262,3264)
- SDP (RFC 2327)
- REFER (RFC 3515)
- RTP/RTCP (RFC 1889,1890)
- STUN (RFC 3489)
- ARP/RARP (RFC 826/903)
- SNTP (RFC 2030)
- DHCP/PPPoE
- TFTP/HTTP/HTTPS
- DNS/DNS SRV (RFC 1706/RFC 2782)
- VLAN 802.1P/802.1Q
- Diff Serve

1.5.2 Voice and Fax parameters

- G.711A/U law, G.723.1, G.729AB
- Comfortable Noise Generation (CNG)
- Voice Activity Detection (VAD)
- Echo Cancellation (G.168)
- Adaptive Dynamic Jitter Buffer
- Voice and fax gain control
- Hook flash Detect
- T.38/Pass-through
- DTMF Mode: Signal/RFC2833/INBAND

1.5.3 Supplementary service

- Busy tone detection
- No current take out stitches detection
- Voice interrupted detection
- One stage dialing
- Two stage dialing
- PSTN exterior ports polling
- Polarity Reversal
- FAS (Fake billing correction)
- DC/AC impedance config
- Calls detection (Bellcore Type 1&2, ETSI,DTMF)

- Direct IP Call
- Primary and secondary SIP account
- 32 inbound/outbound routing
- Number manipulation
- Dial plan set
- Life line

2. Basic Operations

2.1 Phone Call

2.1.1 Phone or Extension Number

- 1) Dial the number directly and wait for 3 seconds (Default "*No dial timeout*");
- 2) Dial the number directly and press #.

2.1.2 Direct IP Calls

DAG series device with FXS port allow two parties directly call through IP address. The user need only a simulation with the FXS port unit equipment linked together and set up calls not registered.

Elements necessary to completing a direct IP call:

- 1) Both DAG serial and other VoIP Device, have public IP addresses;
- 2) Both DAG serial and other VoIP Device are on the same LAN using private IP addresses;
- 3) Both DAG serial and other VoIP Device can be connected through a router using public or private IP addresses (with necessary port forwarding or DMZ).

Operation Process:

- 1) Pick up the analog phone then dial "*47"
- 2) Enter the target IP address.

【Note】: No dial tone will be played between step 1 and step 2

Examples:

If the target IP address is 192.168.0.160, the dialing convention is ***47**, then **192*168*0*160**. Followed by pressing the "#" key or wait 3 seconds. Complete signaling interactive soon after, he was called the unit can be heard ringing.

【Note】: You cannot make direct IP calls between FXS0 to FXS1 since they are using same IP. It only supports the default destination port 5060.

2.2 Call Hold

Place a call on hold by pressing the "flash" button on the analog phone (if the phone has that button). Press the "flash" button again to release the previously held Caller and resume conversation. If no "flash" button is available, use "hook flash" (toggle on-off hook quickly). You may drop a call using hook flash.

2.3 Call Waiting

Call waiting tone (3 short beeps) indicates an incoming call, if the call waiting feature is enabled. Toggle between incoming call and current call by pressing the "flash" button. First call is placed on hold. Press the "flash" button to toggle between two active calls.

2.4 Call Transfer

2.4.1 Blind Transfer

Blind transfer used to transfer call to the third party without inform caller. Assume that call Caller A and B are in conversation. A wants to Blind Transfer B to C:

- 1) Caller A presses **FLASH** on the analog phone to hear the dial tone;
- 2) Caller A dials ***87** then dials caller C's number, and then # (or wait for 4 seconds);
- 3) Caller A will hear the confirm tone. Then, A can hang up.

Note:

"*Call features enable*" must be set to "Yes" in web configuration page. Caller A can place a call on hold and wait for one of three situations:

- 1) A quick confirmation tone (similar to call waiting tone) followed by a dial-tone. This indicates the transfer is successful. At this point, Caller A can either hang up or make another call.
- 2) A quick busy tone followed by a restored call (on supported platforms only). This means the transferee has received a 4xx response for the INVITE and we will try to recover the call. The busy tone is just to indicate to the transferor that the transfer has failed.
- 3) Continuous busy tone. The phone has timed out.

2.4.2 Attended Transfer

Attended transfer allows users to confirm the third party response and decide whether to answer the calls and then transfer this call to the third party.

Assume that Caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:

- 1) Caller A presses **FLASH** on the analog phone for dial tone;
- 2) Dial Caller C's number followed by # (or wait for 3 seconds);
- 3) If Caller C answers the call, Caller A and Caller C are in conversation. Then A can hang up to complete transfer;
- 4) If Caller C does not answer the call, Caller A can press "flash" to resume call with Caller B.

2.5 Call Features

DAG (FXS) support all traditional and senior phone function.

Table 2.5-1 Feature Codec

Feature Codec	Operation Instructions
*158#	View the LAN port IP address
*159#	View the WAN port IP address
*114#	Inquire port account
150	Set the way of obtain IP address

157	Set network method
152	Set IP address
153	Set Subnet mask
156	Set default gateway IP address
*193#	Obtain IP address through DHCP again
*160*1#	Open WAN port to access web
*166*000000#	Factory reset
*111#	Restart device
*#	Call hold
47	IP address call
*51#	Enable call waiting
*50#	Disable call waiting
87	Blind transfer
72	Enable Unconditional Call Forward
*73#	Disable Unconditional Call Forward
90	Enable Busy Call Forward
*91#	Disable Busy Call Forward
92	Enable No Answer Call Forward
*93#	Disable No Answer Call Forward
*78#	Enable DND
*79#	Disable DND
*200#	Access Voice mail
Flash/Hook	Switch between incoming calls, If not in session, flash/hook will switch a new channel for new call.

2.6 Sending and Receiving Fax

2.6.1 DAG (FXS) support four fax modes:

- 1) T.38 (FoIP)
- 2) Pass-Through
- 3) Modem
- 4) adaptive

2.6.2 T. 38 and Pass-Through

T.38 is the preferred method because it is more reliable and works well in most network conditions. If the service provider supports T.38, please use this method by selecting T.38 as fax mode (default). If the service provider does not support T.38, pass-through mode may be used. If you have problems with sending or receiving Fax, toggle the Fax Tone Detection Mode setting.

3. Local IVR Operation

3.1 Inquire IP address

Analog phone connected with FXS ports of device, then pick up, after dial tone, dialing *158# to inquire LAN port IP address and dialing *159# to inquire WAN port IP address.

3.2 Factory Reset

After picking up, dial *166*000000#, then onhook and restart after "Setting successful".

3.3 Configure LAN Port's IP Address

Before configuration, please ensure: (1) The device is power on; (2) devices connecting

to network; (3) Telephone is connecting to FXS port of device.

- 1) Configure dynamic IP address by DHCP:

Offhook; Dial “*150*2#”; Onhook;

If the equipment hint success, after 10 seconds, and restart the equipment.(Power-off then power-on)

- 2) Configure Static IP address

Offhook; Dial “*150*1#”; Onhook;

Then configure IP and mask as follow:

- Configure IP address:

Offhook; input “*152*172*16*0*100#”; onhook

- Configure subnet mask:

Offhook; input “*153*255*255*0*0#”; onhook

- Configure gateway IP address

Offhook; input “*156*172*16*0*1#”; onhook.

- 3) Query the IP address of device: Offhook, input“*158#”

- 4) If the DAG serial uses PPPoE method to get IP address, it need to configure by web browser.

【Note】: the telephone will play voice prompt “Setting successfully” if the step is correct

4. WEB Configuration

4.1 WEB Login

Device is connecting to network properly, refer to chapter 3 “Operation”. Offhook and dial*158# to inquire device IP address.

4.1.1 Login

Device LAN port default IP address is 192.168.11.1, WAN port default obtain IP address by DHCP. Advice to modify the IP address of the local computer equipment and ensure that

are on the same IP segment, with Windows 7 as an example, the local computer IP address change for 192.168.11.10:

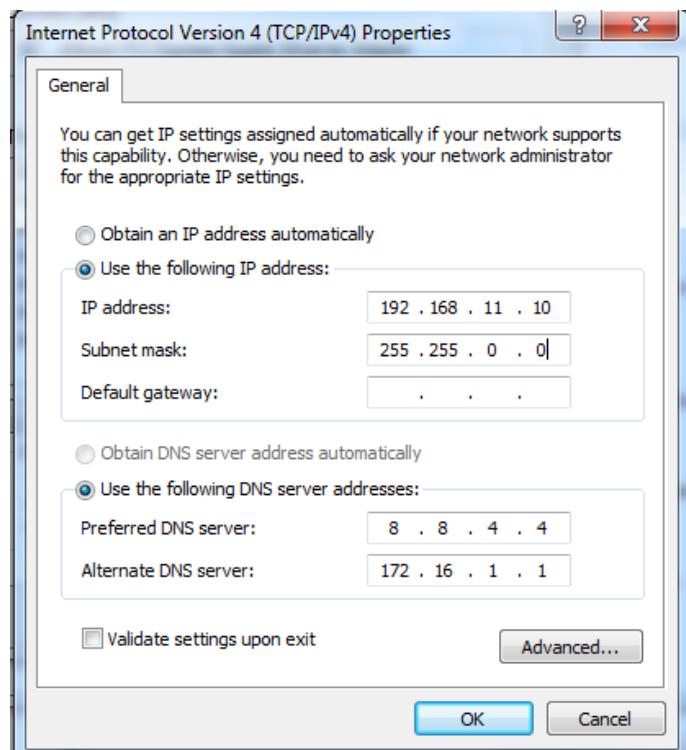


Figure 4.1-1Modify IP address

Check connection between computer and device, click "Start"-> "run"-> input "cmd", run ping 192.168.11.10 –t order to check the connectivity between them.

4.1.2 Login WEB

Open web browser, then input IP address of device, Press "Enter", it pop up logging on identity authentication interface.

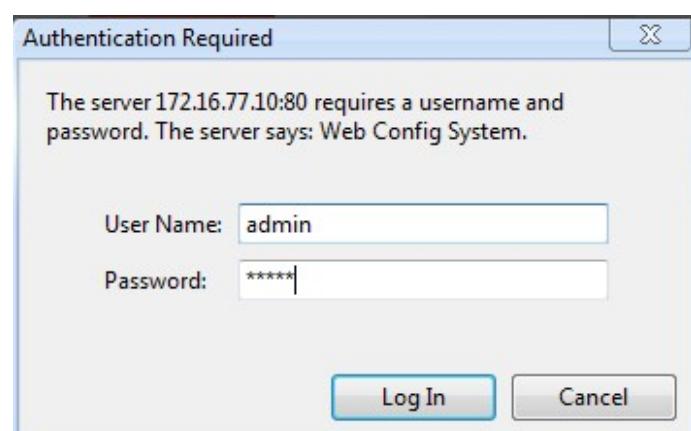


Figure 4.1-1 DAG FXS Login Interface

Default username and password: admin/admin, click "OK" to entry into web interface.

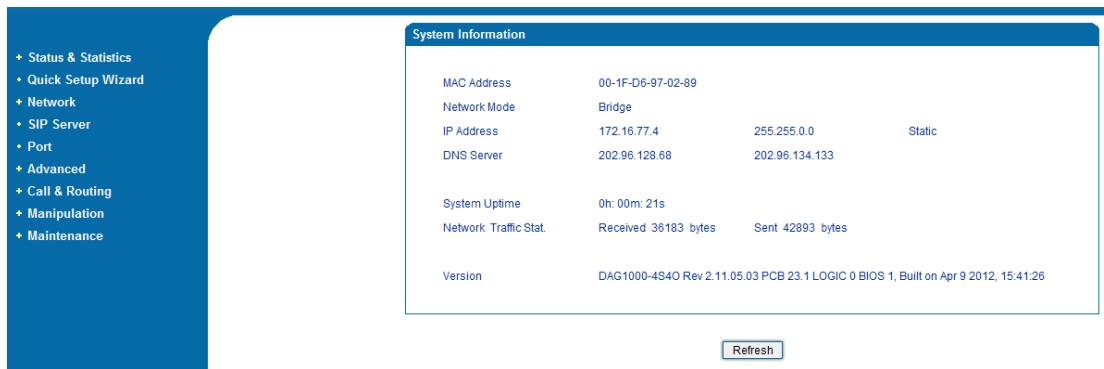


Figure 4.1-2 DAG Configure Interface

4.2 Navigation Tree

DAG series voice gateway web configuration interface mainly includes navigation tree and the right configuration interface. Choose navigation tree in order to entry into the configuration interface.



Figure 4.2-1 Navigation Tree

When device is in bridge mode, navigation tree won't display "routing configuration" items

and the following "DHCP service", "DMZ host", "forward rules" and "static routing" and "ARP" etc.

4.3 State and Statistics

4.3.1 System Information

System information interface shows the run information as following figure 4.3.1 below:

System Information			
MAC Address	00-1F-D6-97-02-89		
Network Mode	Bridge		
IP Address	172.16.77.4	255.255.0.0	Static
DNS Server	202.96.128.68	202.96.134.133	
System Uptime	0h: 05m: 44s		
Network Traffic Stat.	Received 859178 bytes	Sent 257967 bytes	
Version	DAG1000-4S4O Rev 2.11.05.03 PCB 23.1 LOGIC 0 BIOS 1, Built on Apr 9 2012, 15:41:26		
<input type="button" value="Refresh"/>			

Figure 4.3-1(1) System Information (Bridge mode)

System Information			
MAC Address	00-1F-D6-97-02-89		
Network Mode	Router		
WAN IP Address	172.16.77.4	255.255.0.0	Static
LAN Port	192.168.11.1	255.255.255.0	
DNS Server	202.96.128.68	202.96.134.133	
System Uptime	0h: 23m: 24s		
Network Connection Occupancy Ratio	0 %		
WAN Traffic Stat.	Received 3427498 bytes	Sent 1622567 bytes	
Version	DAG1000-4S4O Rev 2.11.05.03 PCB 23.1 LOGIC 0 BIOS 1, Built on Apr 9 2012, 15:41:26		
<input type="button" value="Refresh"/>			

Figure 4.3-1(2) System Information (Router mode)

System information (Router mode) as follow:

Table 4.3-1 System Information Description

MAC address	WAN port hardware address. The device ID in HEX format.
Network Mode	Display network mode, include bridge and router. If it is bridge, WAN port display Network, and the WAN port as same as the LAN port.
WAN Port	<p>Shows WAN IP address of DAG , DHCP mode: all the field values for the Static IP mode are not used (even though they are still saved in the Flash memory.) The DAG acquires its IP address from the first DHCP server it discovers from the LAN it is connected.</p> <p>Using the PPPoE feature: set the PPPoE account settings. The DAG will establish a PPPoE session if any of the PPPoE fields is set.</p> <p>Static IP mode: configure the IP address, Subnet Mask, Default Router IP address, DNS Server 1 (primary), DNS Server 2 (secondary) fields. These fields are set to zero by default.</p>
LAN Port	Shows LAN IP address of DAG. if network Mode is bridge, LAN port won't display.
DNS Server	Display DNS server IP address and default gateway information
System Uptime	Time elapsed from device power on to now.
Network Connection Occupancy Ratio	The current NAT mapping/maximum number of concurrent mapping number.
Network Traffic Statics	Total bytes of message received and sent by network port.
Version	Includes: product mode, software version, hardware version and built time etc.

4.3.2 Registration Information

Port Registration Information					
Port No.	Type	Primary User ID	Primary User Status	Secondary User ID	Secondary User Status
0	FXS	--	--	--	--
1	FXS	--	--	--	--
Port Group Registration Information					
Port Group	Port	Primary User ID	Primary User Status	Secondary User ID	Secondary User Status
--	--	--	--	--	--
--	--	--	--	--	--

[Refresh](#)

Figure 4.3-2 Port and Port group registration information

4.3.3 TCP/UDP Statistics

TCP/UDP Traffic			
TCP Sent Packets	TCP Recv Packets	UDP Sent Packets	UDP Recv Packets
232	59	41	216
<input type="button" value="Refresh"/>			

Figure 4.3-3 TCP/UDP Statistics Information

Figure 4.3-3 shows TCP sending and receiving, UDP sending and receiving packets of statistical information since the device launched.

4.3.4 RTP Session Statistics

RTP Session										
Port	Payload Type	Packet Period	Local Port	Peer IP	Peer Port	Sent Packets	Recv Packets	Lost Packets	Jitter	Duration(s)
--	--	--	--	--	--	--	--	--	--	--
<input type="button" value="Refresh"/>										

Figure 4.3-4 RTP Session Statistics

Figure 4.3-4 display real-time RTP conversation flow data information, includes:

Port, voice codec, packet period, local port, peer IP, peer port, sent packets, receive packets, lost packets, jitter and duration.

4.4 Quick Setup Wizard

Quick configuration guide will guide users to configure the device step by step. Users only need to configure network, SIP server and sip port in quick setup wizard. Basically, after these three steps, users are able to make voice call through device.

4.5 Network Configuration

4.5.1 Local Network

DAG FXS/FXO hybrid gateway has two kinds of work mode: route and bridge. When DAG is set route mode, the DAG will work as small router and NAT function has enabled. In this

situation, WAN port is normally connect to uplink router/switch or ADSL MODEM, LAN port used to connect local computer or other network device(such as Ethernet switches, Hubs etc); When DAG is set bridge mode, WAN and LAN port are the same. The DAG just work as two ports or four ports Ethernet switch.

When it set to bridge mode, only need to configure WAN port IP address and DNS. If set to route mode, default LAN port IP will display and it can be change by users. Network configure interface as below:

Local Network

Network Mode	<input checked="" type="radio"/> Route <input type="radio"/> Bridge
WAN Port	
Link Speed & Duplex	<input type="button" value="Auto Detect"/>
<input type="radio"/> Obtain an IP address automatically	
<input checked="" type="radio"/> Use the following IP address	
IP Address	172.16.77.4
Subnet Mask	255.255.0.0
Default Gateway	172.16.1.5
<input type="radio"/> PPPoE	
Account	
Password	
Service Name	
LAN Port	
Link Speed & Duplex	<input type="button" value="Auto Detect"/>
IP Address	192.168.11.1
Subnet Mask	255.255.255.0
DNS Server	
<input type="radio"/> Obtain DNS server address automatically	
<input checked="" type="radio"/> Use the following DNS server address	
Primary DNS Server	202.96.128.68
Secondary DNS Server	202.96.134.133
<input type="button" value="Save"/>	

Figure 4.5-1Route Mode

Local Network

Network Mode Route Bridge

Network Configuration

Link Speed & Duplex	<input type="text" value="Auto Detect"/> <input type="button" value="▼"/>
<input type="radio"/> Obtain an IP address automatically	
<input checked="" type="radio"/> Use the following IP address	
IP Address	<input type="text" value="172.16.77.4"/>
Subnet Mask	<input type="text" value="255.255.0.0"/>
Default Gateway	<input type="text" value="172.16.1.5"/>
<input type="radio"/> PPPoE	
Account	<input type="text"/>
Password	<input type="text"/>
Service Name	<input type="text"/>

DNS Server

<input type="radio"/> Obtain DNS server address automatically	
<input checked="" type="radio"/> Use the following DNS server address	
Primary DNS Server	<input type="text" value="202.96.128.68"/>
Secondary DNS Server	<input type="text" value="202.96.134.133"/>

Note: The device must restart to take effect.

Figure 4.5-2 Bridge Mode

- “Link Speed & Duplex” used to select Ethernet port work mode, include 5 kinds of choice, “Auto Detect”、 “10Mbps half-duplex”、 “10Mbps full-duplex”, “100Mbps half-duplex”, “100Mbps full-duplex”, default is “Auto Detect”.
- When select “Obtain IP address automatically”, DAG will obtain IP address by DHCP.
- When select “Use the following IP address”, that configure DAG to fixed IP address mode.
- When select “PPPoE”, please fill in account and password offered by ISP in internet account and password.

【Notes】:

- If select automatically obtain IP address, please ensure DHCP server in network and work normally.

- 2) Under route mode, please configure LAN port and WAN port in different segment, otherwise DAG can't work normally.
- 3) Under route mode, login DAG configuration interface only used LAN port.
- 4) After configuration, restart device configuration validation.

4.5.2 VLAN Parameter

Generally, Internet provides only Best Effort Service. Since Ethernet is the most spread LAN access technology, importance of providing it a quality of service mechanism ought not to be neglected.

Ethernet technology also used as WAN technology, not only as LAN technology. Due to rapidly increasing use Internet through Public Switched Telecommunication Network (PSTN), Telephone Companies are forced to implement IP-based networks as their PSTN backbones. A network like this without any Quality of Service mechanisms would be disastrous. Just imagine yourself trying to get an emergency call through while others just surf the Internet.

1) 802.1Q

The IEEE 802.1Q standard defines architecture for Virtual Bridged LANs, the services provided in Virtual Bridged LANs and the protocols and algorithms involved in the provision of those services.

No Quality of Service mechanisms are defined in this standard, but an important requirement for providing QoS is included in this standard, e.g. ability to regenerate user priority of received frames using priority information contained in the frame and the User Priority Regeneration Table for the reception Port.

2) 802.1p

IEEE 802.1p standard, Traffic class expediting and dynamic multicast filtering. It describes important methods for providing QoS at MAC level. IEEE 802.1p is in fact quite good. Lower priority level packets are not sent, if there is packets in queued in higher level

queues. IEEE 802.1p describes no admission control protocols. It would be possible to give Network Control priority to all packets and the network would be easily congested.

There are three VLAN: data VLAN, voice LAN and management VLAN. VLAN configuration interface as following figure 4-4-3:

VLAN

Data VLAN

Enable
0

Data 802.1Q VLAN ID (0 - 4095)
0

Data 802.1P Priority (0 - 7)
0

In this case, data VLAN uses the default WAN interface.

Voice VLAN

Enable
0

Voice 802.1Q VLAN ID (0 - 4095)
0

Voice 802.1P Priority (0 - 7)
0

Voice VLAN uses following separate IP interface.

Obtain an IP address automatically
 Use the following IP address

IP Address
Subnet Mask
Default Gateway

Management VLAN

Enable
0

Management 802.1Q VLAN ID (0 - 4095)
0

Management 802.1P Priority (0 - 7)
0

Management VLAN uses following separate IP interface.

Obtain an IP address automatically
 Use the following IP address

IP Address
Subnet Mask
Default Gateway

Save

Figure 4.5-3 VLAN parameter configuration

Table 4.5-1VLAN parameter configuration

Data VLAN	Data 802.1Q VLAN ID(0-4095)	Fill out an ID to describe a data VLAN group, ID 0 used to management VLAN, can't used to service configure.
	Data 802.1p Priority (0-7)	802.1 protocol to control network traffic priority, Priority from 0-7.

Voice VLAN	Voice 802.1Q VLAN ID(0-4095)	Fill out an ID to describe a voice VLAN group, ID 0 used to management VLAN, can't used to service configure.
	Voice 802.1p Priority (0-7)	802.1 protocol to control network traffic priority, Priority from 0-7.
	IP address	Can use dynamic or static IP address
	Voice VLAN DNS Server	Can use dynamic or static DNS server address
Management VLAN	Management 802.1Q VLAN ID(0-4095)	Fill out an ID to describe a data VLAN group, ID 0 used to management VLAN, can't used to service configure.
	Management 802.1p Priority (0-7)	802.1 protocol to control network traffic priority, Priority from 0-7.
	IP address	Can use dynamic or static IP address
	Management VLAN DNS server	Can use dynamic or static DNS server address

【Note】: restart the device to take configuration effect.

4.5.3 MAC Clone (Routing mode)

MAC Clone

This page provides the setting MAC address of WAN

PC MAC Address:	BC-AE-C5-4A-79-E9	Clone
Device MAC Address:	00-1F-D6-97-02-7D	Restore

Save

Note:The device must restart to take effect.

Figure 4.5-4 MAC Clone Interface

More client in LAN have already can't share internet used the traditional "gateway set law".

Because IP address binding in only a legitimate MAC address by ISP. If the ISP's switch discovers illegal MAC address, it will refuse the service.

The best way is MAC clone for MAC binding. Most ADSL MODEM, broadband router, wireless router have this feature. The principle of MAC address clone is deliberately exposed MAC address of bound computer to the ISP server and let the ISP server think that used only a single piece of computer, in fact many computers in sharing the Internet.

This function used to prevent ISP limiting to share the Internet.

【Note】: Restart device to take configuration effect.

4.5.4 DHCP Server (Routing mode)

Under route mode, DAG network part as a small router to configure DHCP service, that DAG as a DHCP server in network.

Start and end address of address pool determine the range of IP address automatically assigned to other devices;

- IP Expire Time means use time of assigned IP address. More than the lease time, if the IP address is not used by network equipment, IP address will be recovered;
- Subnet mask, gateway, DNS and other information configured by DHCP protocol.

Configuration interface as figure 4.5-5:

DHCP Server Config	
DHCP Server	<input type="checkbox"/> Enable
IP Pool Starting Address	192.168.11.100
IP Pool Ending Address	192.168.11.199
IP Expire Time	72 h
Subnet Mask (Optional)	255.255.255.0
Default Gateway (Optional)	192.168.11.77
Primary DNS Server (Optional)	202.96.128.68
Secondary DNS Server (Optional)	202.96.134.133

Note:The device must restart to take effect.

Figure 4.5-5 DHCP Configuration Interface

【Note】: When configure start and end IP address, subnet mask and gateway, please set the same segment with LAN port. Otherwise, device will not work normally. After configuration, restart device configuration validation.

4.5.5 DMZ Host (Routing mode)

DMZ (Demilitarized Zone) connect web, e-mail etc. server allowed external to access to this area. Make the internal network located the back of the zone of confidence and not allow any access, separation of inside and outside the network, protect user

information. DMZ can be understood that a special areas of the network and different from the external network or intranet. Public server that does not contain confidential information usually placed in DMZ, such as web, Mail, FTP etc. Accuser from intranet can visit the service of DMZ, but can't come into contact with confidential or private information stored in the network. Even if DMZ server is damaged, it will not be confidential information in the internal network.

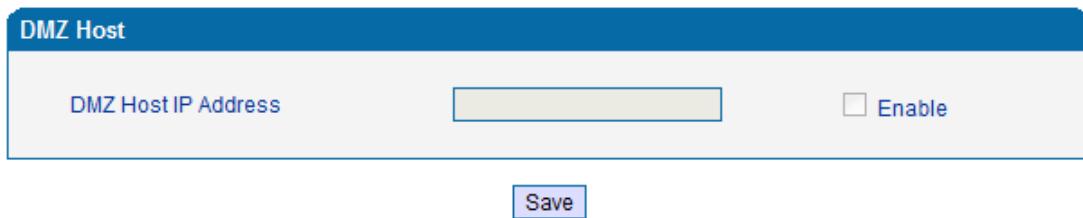


Figure 4.5-6 DMZ Configuration Interface

【Note】: After configuration, restart device configuration validation.

4.5.6 Forward Rule (Routing mode)

In some cases, LAN network equipment need to provide some communication in WAN network (such as port for 21 FTP service), This time can be configured forwarding rules for the network equipment.

Service ports namely the need to provide service network mouth WAN ports, IP address that LAN network provide services to the mouth of the network equipment IP address, the protocol is TCP or UDP.

The different between forward rule and DMZ host is that DMZ Host offers continuous multiple

Port (0-1024) and all the foreign communication agreement; while the forward rule offers a single or a few port foreign communication on some protocol. When the conflicts exist between forward rule and DMZ host, the configuration of forwarding rules is preferred.

Forward rule configuration interface as follows:

Forward Rule Table

ID	Server Port	IP Address	Protocol	Enable
1			TCP	<input type="checkbox"/>
2			TCP	<input type="checkbox"/>
3			TCP	<input type="checkbox"/>
4			TCP	<input type="checkbox"/>
5			TCP	<input type="checkbox"/>
6			TCP	<input type="checkbox"/>
7			TCP	<input type="checkbox"/>
8			TCP	<input type="checkbox"/>

Save

Notes: (1) 'IP Address' needs to be in the same subnet with LAN port.
(2) 'Server Port' range: 0 - 65535.

Figure 4.5-7 Forward rule configuration interface

4.5.7 Static Route Table

Static Route Table is IP communication direction in network, generally do not need to configure static route. When there are many segments in LAN network and need to complete some specific application among these segments, the static route need to be configured.

Static Route configuration interface as follows:

Static Route Table

ID	Dest. IP Address	Subnet Mask	Nexthop	Enable
1				<input type="checkbox"/>
2				<input type="checkbox"/>
3				<input type="checkbox"/>
4				<input type="checkbox"/>
5				<input type="checkbox"/>
6				<input type="checkbox"/>
7				<input type="checkbox"/>
8				<input type="checkbox"/>

Save

Figure 4.5-8 Static route configuration Interface

4.5.8 ARP

ARP brief introduction:

ARP is address resolution protocol. After configuring ARP, users can get physical address through device IP address. Under TCP/IP network environment, each host is assigned a 32-bit IP address. But the message transmission needs to know the purpose the physical address of the party. ARP is a tool that converts IP address into MAC address.

ARP configuration interface as follows:

Type	IP Address	MAC Address
<input checked="" type="radio"/> Static <input type="radio"/> Dynamic	—	—

Total: 0

Figure 4.5-9 ARP Parameters

4.6 SIP Server

SIP server introduction:

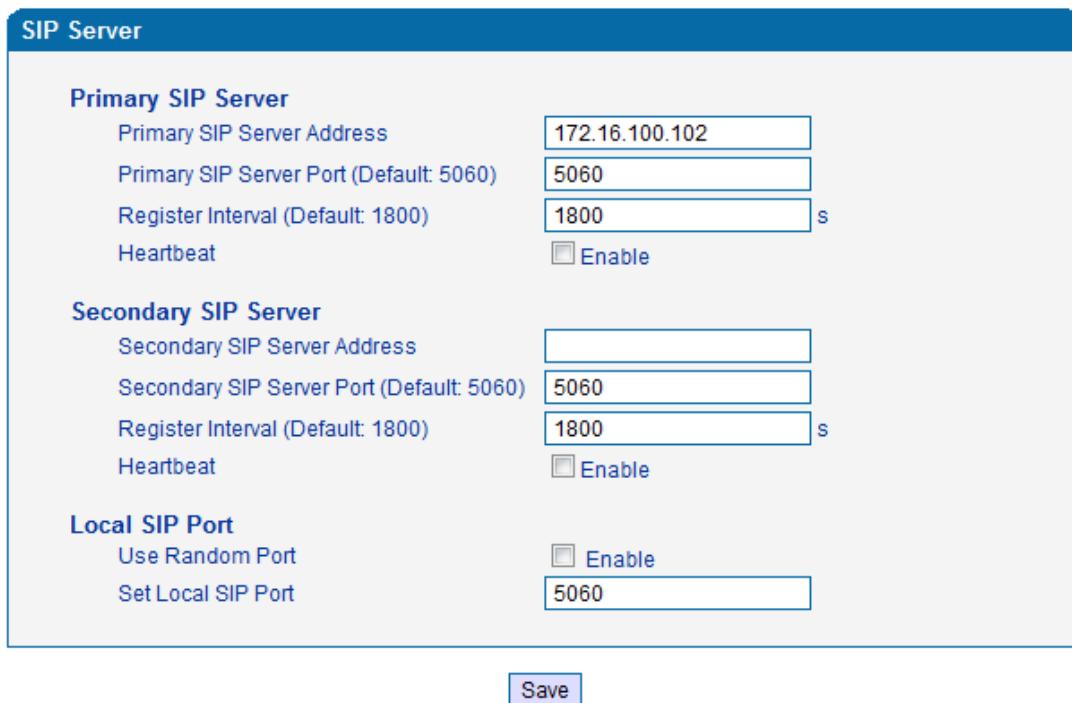
- 1) SIP server is the main component of VoIP network and responsible for establishing all the SIP phone calls. SIP server also called SIP proxy server or registered server. IPPBX and the soft-switch can act as SIP server role.
- 2) Usually, SIP server does not participate in the media process.

In SIP network, the media always using end-to-end to hand the consultation. In some particular situation or business processing, such as "Music On Hold", SIP server will actively participate in the media negotiation. Simple SIP server is responsible only for establishment, maintenance and cleaning conversation, don't interfere in call. While relatively complex SIP server also called SIP PBX. It not only provides the basic call, and basic conversational support, also offer plenty of business, such as: Presence, Find-me, Music On Hold.

- 3) SIP server based on Linux platform, such as: OpenSER, sipXecx, VoS, Mera etc.
- 4) SIP server based on windows platform, such as :miniSipServer, Brekeke, VoIPswitch etc.

5) Carrier grade soft-switch platform, such as Cisco, Huawei, Zteetc.

SIP server configuration interface as follows:



SIP Server

Primary SIP Server

Primary SIP Server Address: 172.16.100.102

Primary SIP Server Port (Default: 5060): 5060

Register Interval (Default: 1800): 1800 s

Heartbeat: Enable

Secondary SIP Server

Secondary SIP Server Address:

Secondary SIP Server Port (Default: 5060): 5060

Register Interval (Default: 1800): 1800 s

Heartbeat: Enable

Local SIP Port

Use Random Port: Enable

Set Local SIP Port: 5060

Save

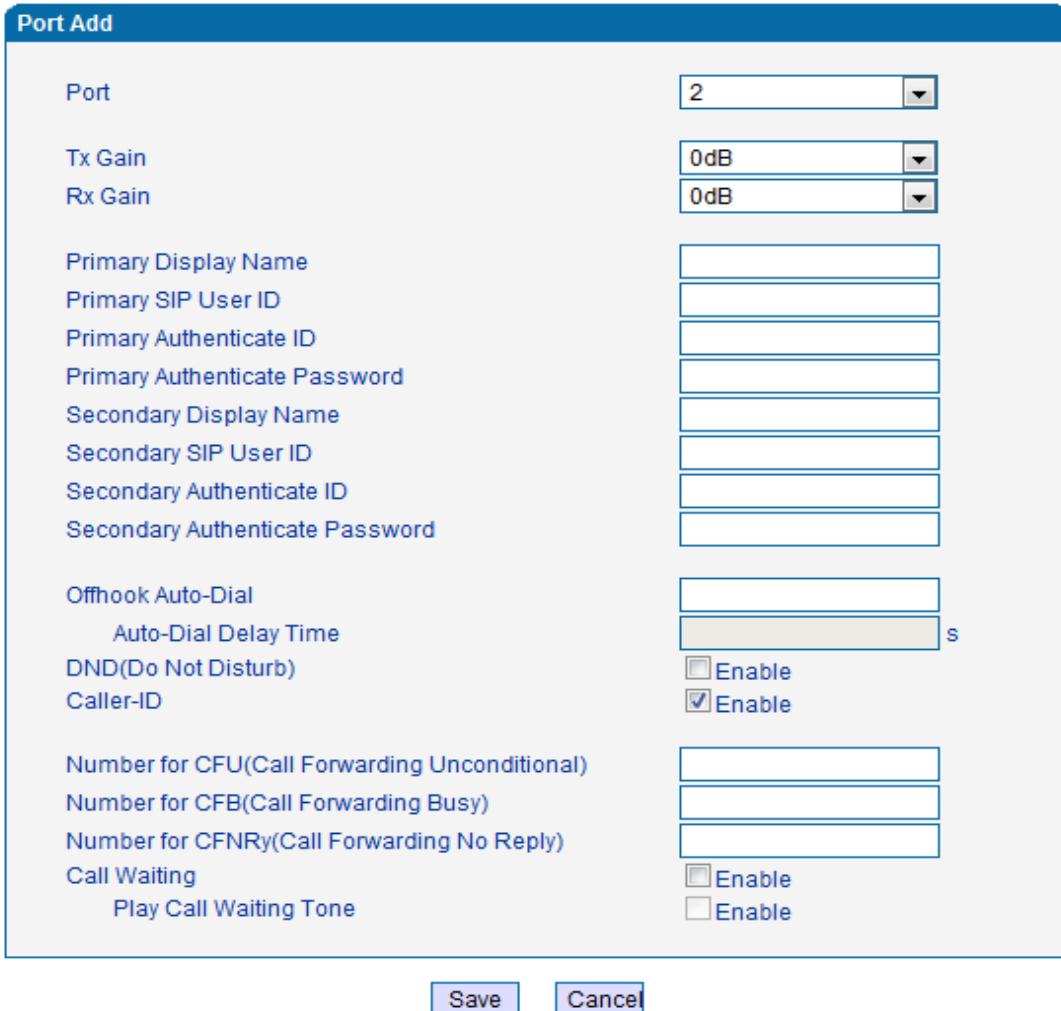
Figure 4.6-1 SIP Server Configuration Interface

SIP parameter description:

Primary SIP Server IP	SIP Server IP address or Domain name provided by VoIP service provider.
Primary SIP Server port	Service port, default is 5060
Register interval	protects registrar against excessively frequent registration refreshes while limiting the state. Every once in a while send request for registration to the terminal server, default is 1800s.
Heartbeat	Heartbeat message detect the connection status between device and SIP server.
Secondary SIP Server IP address	Backup SIP Server's IP address or Domain name provided by VoIP service provider.
Secondary SIP Server port	Service port, default is 5060
Secondary SIP server Register interval	protects registrar against excessively frequent registration refreshes while limiting the state. Every once in a while send request for registration to the terminal server, default is 1800s.
Secondary SIP heartbeat	Heartbeat message detect the connection status between device and SIP server.
Use Random Port	Random SIP service ports for DAG
Set Local SIP port	Default SIP service port is 5060.

4.7 Port Configuration

Port parameters include: Send gain, receive gain, primary display name etc.



The screenshot shows the 'Port Add' configuration interface. It includes fields for Port (set to 2), Tx Gain (0dB), Rx Gain (0dB), and various SIP and authentication parameters for both Primary and Secondary accounts. It also includes sections for Offhook Auto-Dial, DND, and Caller-ID settings, as well as Call Forwarding options (CFU, CFB, CFNRy) and Call Waiting settings. At the bottom are 'Save' and 'Cancel' buttons.

Parameter	Value
Port	2
Tx Gain	0dB
Rx Gain	0dB
Primary Display Name	
Primary SIP User ID	
Primary Authenticate ID	
Primary Authenticate Password	
Secondary Display Name	
Secondary SIP User ID	
Secondary Authenticate ID	
Secondary Authenticate Password	
Offhook Auto-Dial	
Auto-Dial Delay Time	s
DND(Do Not Disturb)	<input type="checkbox"/> Enable
Caller-ID	<input checked="" type="checkbox"/> Enable
Number for CFU(Call Forwarding Unconditional)	
Number for CFB(Call Forwarding Busy)	
Number for CFNRy(Call Forwarding No Reply)	
Call Waiting	<input type="checkbox"/> Enable
Play Call Waiting Tone	<input type="checkbox"/> Enable

Figure 4.7-1 Port configuration interface

Port parameters introduce as follows:

Tx Gain	It is use to control the volume of conversation, Adjust "TX gain" will affect the end users voice size, the default value is 0. Its value range from -10 – 10 dB
Rx Gain	It is use to control the volume of conversation, Adjust "RX gain" will affect the end users voice size, the default value is 0. Its value range from -10 – 10 dB
Primary /Secondary SIP Display Name	Primary /Secondary SIP account description, Its purpose is so you can identify the SIP account with a meaningful name
Primary /Secondary SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Primary/Secondary SIP	SIP service subscriber's Authenticate ID used for authentication. Can be

Authenticate ID	identical to or different from SIP User ID.
Primary/Secondary Authenticate password	SIP password which registers to soft switch/SIP server
Offhook Auto-dial	Pre-assign an extension or phone number so that automatically dial a number as soon as you pick up the phone set
Auto-dial Delay Time	Delay 0-3 seconds to automatically dial a number, 0 means dial number immediately
DND	Do not disturb, the phone set won't receive any calls in case it enabled
Caller ID	Enable or disable caller ID for corresponding port
Number for CFU	call forward unconditional, all incoming calls will forward to pre-assigned number automatically
Number for CFB	Call forward on busy, if the line is busy, the call will forward to pre-assigned number automatically
Number for CFNRy	Call forward no reply, if the line is not answer the call, the call will forward to pre-assigned number automatically
Call Waiting	If call waiting enabled, it will send a special tone if another caller tries to reach you when you are using your telephone
Play Call Waiting Tone	Enable call waiting tone, caller will hear special tone.

4.8 Advanced

4.8.1 FXS/FXO parameters

FXS characteristic parameters include: Call progress Tone, Timeout for Dialing, Send Polarity Reversal etc.

FXO full name is Foreign Exchange Office. It is a kind of voice interface, and a trunk connected between central exchange switches and telephone exchange system. To central office speaking, it simulates a PABX extension, and can realize connection among common phone and a multiplexer. It also is FXO interface connected with SPC exchanges.

FXO as ordinary telephone interface, and need to remote provide current. FXO may connect company's internal PBX service extension and the telecom outside, generally speaking, FXO is a telephone. So just lead a inside to FXO port from company's internal, or directly line a straight up in FXO from the telecom.

FXO parameters include: Call progress Tone, Timeout for Dialing, Send Polarity Reversal

etc. Configuration interface as follow:

FXS / FXO

Call Progress Tone	USA
Timeout for Dialing	4 s
Timeout for Answer(Outgoing Call)	55 s
Timeout for Answer(Incoming Call)	55 s
Life-Line when Registration Failed	<input type="checkbox"/> Enable
FXS Parameter	
Send Polarity Reversal	<input type="checkbox"/> Enable
Detect Hook Flash	<input checked="" type="checkbox"/> Enable
Min Time	100 ms
Max Time	400 ms
CID Type	FSK
Message Type	MDMF
Send CID before Ringing	<input type="checkbox"/> Enable
Delay of Sending CID after Ringing	500 ms
CFNRy Timeout	33 s
SLIC Setting	600 Ohm
FXO Parameter	
Incoming Call from PSTN	
Configuration by FXO	<input checked="" type="checkbox"/> Enable
Detect CID	<input checked="" type="checkbox"/> After Ring
Send Original CID when Call from PSTN	<input type="checkbox"/> Enable
FXO Keep Onhook until Callee Answered	<input type="checkbox"/> Enable
Play Hint to FXO	<input checked="" type="checkbox"/> Enable
Outgoing Call to PSTN	
One Stage Dialing	<input checked="" type="checkbox"/> Enable
Dial Delay	400 ms
Answer to Caller when	
Polarity Reversal Detected	<input type="checkbox"/> Enable
Delay Time after FXO Offhook	5 s
Onhook when	
Busy Tone Detected	<input checked="" type="checkbox"/> Enable
No Current Detected	<input type="checkbox"/> Enable
Current Disconnect Threshold	200 ms
No RTP Detected	<input checked="" type="checkbox"/> Enable
Period without RTP Packet	60 s
DC Impedance	50 Ohm
AC Impedance	600 Ohm

Save

Figure 4.8-1 FXS/FXO Parameters Configuration Interface

FXS/FXO parameters description:

Call Process Tone	Hear the dial tone when pick up the phone. Choose the national standards from the drop-down box. Default is the United States.
Timeout for dialing	With the help of dialing timeout, you can limit the time while users typing the digits from an extension. If the timeout expire while the user is typing in the extension then DAG will consider the extension as complete and it will try to send to SIP server. Default value is 4 seconds
Timeout for answer(Outgoing call)	This timer set how long the caller party waiting when makes outgoing call on extension.
Timeout for answer(Incoming call)	This timer set how long the phone sets ringing when get incoming call
Life-Line when Registration Failed	Under device are not registered circumstance, call inbound directly from corresponding FXO port.
Send Polarity Reversal	Enable polarity reversal to billing.
Detect Hook flash	A protruding button where putting the receiver boards, called Flash. Always press is hang up, pick up the receiver, the fork lift machine from reed called, by hand clap called "Hook flash". Hook flash is a process that put the flash fast by pressing and let go. In essence is to cut off the dc access about 80 to 200ms. Then switches don't think it's hang on, but keep the call, taking some other operating. The typical application of hook flash is the telephone switchboard. When need to transfer the call to other extension, then telephone hook flash to transfer the call.
CID Type	There are DTMF and FSK, General for the default.
Message Type	The call display formats SDMF and MDMF, General for the default
Send CID before Ringing	After enable this configuration, The DAG send caller to phone set before ringing, otherwise the caller ID will display after ringing.
Delay of sending CID after Ringing	Definite delay timer of caller ID while it set to send caller ID after ringing. Its Default value 500ms
CFNRY Timeout	The time of no answer call transfer (This time must less than call in no answer overtime)
SLIC Setting	Set the unit impedance
Configuration by FXO	When the call from FXO interface, users can be enable or disabled FXO allocation function. FXO configuration function includes: detect CID, Send original CID, Play hint to FXO.
Detect CID	FXO detection calling number and the order of the ring. System has two modes: Before ring and After ring.
Send Original CID when Call from PSTN	Enable this function, the extension call display will show the PSTN side number. Otherwise, the call display will show FXO port number.
FXO Keep Onhook until Callee Answered	Enable this function, when call from PSTN to FXO port, FXO port set auto-dial, and pick up after the extension number connection. This function

	mainly used to billing.
Play Hint to FXO	Enable this function, when call from PSTN to FXO port, FXO port will play prompt tone "please dial the extension number".
One Stage Dialing	Enable this function, FXO port directly sent the dial number, without call extension.
Dial Delay	When call from FXO port to PSTN, the interval of sending number by FXO port, default is 400ms.
Polarity Reversal Detected	When call from FXO port to PSTN, the way of FXO response caller is detecting polarity reversal. If device detected polarity reversal, and then reported to caller to respond, and began to billing.
Delay Time after FXO Offhook	The time of responding caller by FXO port should be less than this configuration.
Onhook when Busy Tone Detected	One of the FXO onhook conditions. When FXO port detected busy tone, FXO will onhook.
Onhook when NO Current Detected	Another of the FXO onhook conditions. When FXO port detected no current, FXO will onhook.
Current Disconnect Threshold	Default the time of no current should be less than 200ms.
No RTP Detected	Enable this function, the system will detect whether RTP flow is interrupted. When voice interrupted, this function can prevent FXO port hanged.
Period without RTP Packet	How long time no RTP packet allowed.
AC/DC Impedance	Adjust impedance, used to impedance matching when FXO and PBX docking.

4.8.2 Media Parameter

Media parameter mainly include: RTP start port, DTMF parameter, Preferred Vocoder.

Configuration Interface as follow:

Media Parameter

RTP Start Port	8000			
DTMF Parameter				
DTMF Method	SIGNAL			
DTMF Gain	0dB			
DTMF Send Interval	200 ms			
Preferred Vocoder				
Coder Name	Payload Type	Packetization Time(ms)	Rate(kbps)	Silence Suppression
1st G729	18	20	8	Disable
2nd G711U	0	20	64	Disable
3rd G711A	8	20	64	Disable

Save

Figure 4.8-2 Media Parameter Configuration Interface

Media parameter description:

RTP Start Port	Default RTP port 8000
DTMF Method	SINGAL、INBAND、RFC2833
RFC2833 Payload Type Optimization	It is configurable When RFC2833 is selected, payload negotiation parameter with remote side, it includes two options: Local and remote
RFC2833 Payload Type	Payload value, default is 101
DTMF Gain	Default is 0 DB
DTMF Send Interval	DTMF send signal interval, default is 200ms.
Coder Name	DAG supports G729、G711U、G711A、G723. while it make outgoing call, G.729 will used as figure 4.8.2 displayed
Payload Type	Each kind of coding has a unique type load value, refer toRFC3551
Packetization Time	Voice package time
Rate	Voice data flow rate, system default
Silence Suppression	Default is disable, if enable, according to the current noise environment dynamically adjust mute inhibit threshold, thus in the user in silent state stop transmission background noise bag and save about VoIP bandwidth. In

	the low bandwidth environment, can reduce the network congestion, greatly improving VoIP call effect.
--	---

4.8.3 SIP Parameter

SIP Parameter

SUBSCRIBE for MWI(Message Waiting Indicator)	<input type="checkbox"/> Enable
Voicemail User ID	<input type="text"/>
RTP Mode in SDP when Call Holding	Sendonly
IP-to-IP Call	<input type="checkbox"/> Enable
URI includes "user=phone"	<input type="checkbox"/> Enable
Only Accept Calls from Server	<input type="checkbox"/> Enable
Anonymous Call	<input type="checkbox"/> Enable
Reject Anonymous Call	<input type="checkbox"/> Enable
Send Flash Event	<input type="checkbox"/> Enable
"#" as Ending Dial Key	<input checked="" type="checkbox"/> Enable
PRACK	<input type="checkbox"/> Enable
Value of "Refer To" refers to "Contact"	<input type="checkbox"/> Enable
Domain Query Type	A Query
Domain Re-resolution Interval(0 means disable)	0 min
T1	500 ms
T2	4000 ms
T4	5000 ms
Max Timeout	32000 ms
Heartbeat Interval(1 - 3600s)	10 s

Save

Figure 4.8-3 SIP Parameter Configuration Interface

SIP parameter description:

SUBSCRIBE for MWI	Voicemail message indicator, it is to be realized in the way of NOTIFY
Voicemail User ID	Access code to voicemail box
RTP Mode in SDP when Call Holding	When call come into holding, if select to receive and not send packet, then the local can hear call waiting tone. If select to not receive and not send packet, then doesn't play call waiting tone.
IP-to-IP Call	Enable this function, users may use the * business call IP address on the phone.

URI Includes user=phone	SIP carries the information, the system defaults not open.
Only Accept Call from Server	Default is no, it indicates the DAG accept incoming call from SIP server only
Anonymous Call	Enable anonymous call, "anonymous" will include in SIP message
Reject Anonymous Call	Enable this function, reject all anonymous call. Disable by default
Send Flash Event	After hook flash, flash event will report flash message to server and server deal with this information.
# as ending Dial Key	Dial-up, use # as a end descriptor.
PRACK	RFC3262 defined an optional extension methods—PRACK (provisional ACK), Used to support the reliability of the temporary response.
Value of "Refer To" refers to "Contact"	Its function is to require the receiving party contact with the third party through the use of supplied in the request in the address information. "Refer to" field of SIP message fill in "contact header".
Domain Query Type	There are two modes option: A QUERY and SRV QUERY. Default is A QUERY.
Domain Re-resolution Interval	Default 0: forbidden
T1	T1 timer of SIP protocol, default is 500ms
T2	T2 timer of SIP protocol, default is 400ms
T4	T4 timer of SIP protocol, default is 500ms
Max Timeout	The max timeout of sending or receiving, default is 32s
Heartbeat Interval	Default is 10s.

Voice mail instructions:

Here DAG work with Elastix the example, introduces how voicemail work in DAG.

- 1) DAG register to Elastix server. Corresponding extension number enable voice mail function in Elastix and set password. As below:

Voicemail & Directory

Status	Enabled <input type="button" value="▼"/>
Voicemail Password	111111
Email Address	<input type="text"/>
Pager Email Address	<input type="text"/>
Email Attachment	<input type="radio"/> yes <input checked="" type="radio"/> no
Play CID	<input type="radio"/> yes <input checked="" type="radio"/> no
Play Envelope	<input type="radio"/> yes <input checked="" type="radio"/> no
Delete Voicemail	<input type="radio"/> yes <input checked="" type="radio"/> no
IMAP Username	<input type="text"/>
IMAP Password	<input type="text"/>
VM Options	<input type="text"/>
VM Context	default
VmX Locator	<input type="text"/>

Figure 4.8-4 Elastix Voicemail Configuration Interface

2) Check feature code in Elastix and change it as necessary. Its default feature codes setting as below:

Voicemail

Dial Voicemail	*98	<input checked="" type="checkbox"/>	Enabled <input type="button" value="▼"/>
My Voicemail	*97	<input checked="" type="checkbox"/>	Enabled <input type="button" value="▼"/>

Figure 4.8-5 Elastix Voicemail Setting

SIP Parameter

SUBSCRIBE for MWI(Message Waiting Indicator)	<input type="checkbox"/> Enable
Voicemail User ID	<input type="text"/>

Figure 4.8-6 Voice Mail Setting In SIP Parameter

3) Enable voice mail in DAG and Elastix will ask you to leave a message after ringing 15 seconds, then Elastix will record and display your message.

Voicemail

Ringtime Default:

Direct Dial Voicemail Prefix:

Direct Dial to Voicemail message type:

Optional Voicemail Recording Gain:

Do Not Play "please leave message after tone" to caller

Figure 4.8-7 Voicemail Setting

4) DAG dial *200#, then dial voicemail account and then ask password for Validation. After that the user will hear voice message.

4.8.4 Fax Parameter

Fax introduction:

DAG fax parameter includes: fax mode, Fax sound detection party, ECM, Rate.

Fax Config

Mode	<input type="text" value="T.38"/>
Tone Detection by	<input type="text" value="Auto"/>
ECM Enable	<input type="checkbox"/>
Rate	<input type="text" value="14400 bps"/>

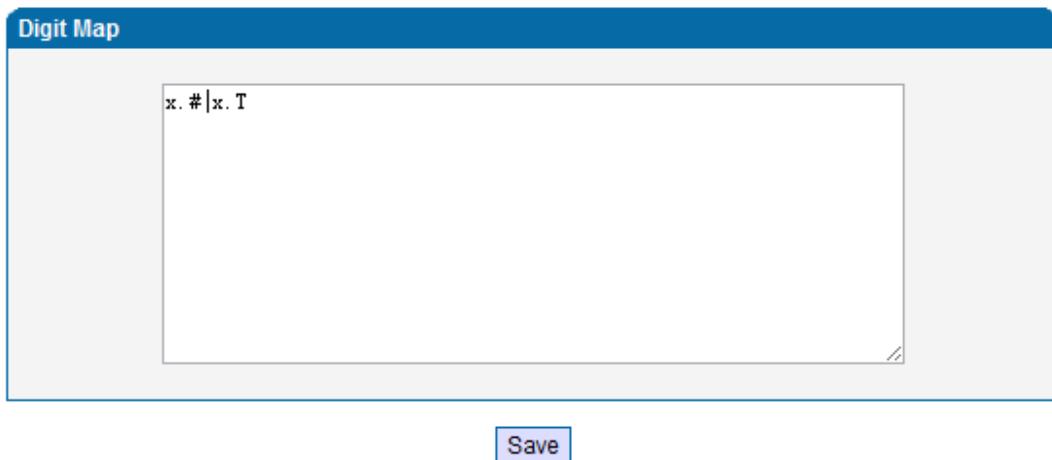
Save

Figure 4.8-8 Fax Parameter Configure Interface

Fax parameter description:

Mode	Fax mode support T.38, T.30(Pass-through),Modem, Adaptive.
Tone Detection by	Fax sound detection mode: Caller, Callee, Automatic.
ECM	Fax error correction information
Rate	The rate of sending and receiving.

4.8.5 Digit Map



NOTE: Length of 'Digit Map' should not be more than 119 characters.

Figure 4.8-9Digit Map

Gateway is collect digits dialed by user, if received a number to be immediately report, the efficiency is too low and a large number of take up network resources. A reasonable method is concentration sending a message after receiving all number. How to judge the gateway receiving all number is the difficulties of this method. The solution is the call agent loading a "Digit Map" to gateway.

Digit Map includes a series figure characters, when the dial-up sequence and one received a character string matching, it means the number has received neat. Digital string contains characters allowed: data0~9, letterA~D, "#", "*", letter T, letter x and ". ". "|" parts of each string is a choice of dial-up solutions; "[]" means choose anyone; "*" means one reports; letter T means detected timer overtime; x means any data; "." means multiple characters can be behind, include 0; "#" means report immediately.

Digit Map Syntax:

1. Supported objects

Digit: A digit from "0" to "9".

Timer: The symbol "T" matching a timer expiry.

DTMF: A digit, a timer, or one of the symbols "A", "B", "C", "D", "#", or "*".

2. Range []

One or more DTMF symbols enclosed between square brackets ("[" and "]"), but only one can be selected.

3. Range ()

One or more expressions enclosed between round brackets ("(" and ")"), but only one can be selected.

4. Separator

|: Separated expressions or DTMF symbols.

5. Subrange

-: Two digits separated by hyphen (" - ") which matches any digit between and including the two. The sub range construct can only be used inside a range construct, i.e., between "[" and "]".

6. Wildcard

x: matches any digit ("0" to "9").

7. Modifiers

.: Match 0 or more times.

8. Modifiers

+: Match 1 or more times.

9. Modifiers

? : Match 0 or 1 times.

Example:

Assume we have the following digit maps:

1. xxxxxxx | x11

and a current dial string of "41". Given the input "1" the current dial string becomes "411". We have a partial match with "xxxxxx", but a complete match with "x11", and hence we send "411" to the Call Agent.

2. [2-8] xxxxxx | 13xxxxxxxx

Means that first is "2","3","4","5","6","7" or "8", followed by 6 digits;

or first is 13, followed by 9 digits.

3. (13 | 15 | 18)xxxxxxxx

Means that first is "13","15" or "18", followed by 8 digits.

4. [1-357-9]xx

Means that first is "1","2","3" or "5" or "7","8","9", followed by 2 digits.

4.8.6 Feature Codec

Feature codec includes device function and call function. Feature codec as follow:

Feature Code			
Feature	Codes	Use Default	Status
Device Function			
Inquiry LAN IP	*158#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Inquiry WAN IP	*159#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Inquiry Phone Number	*114#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Setting IP Mode	*150*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Network Work Mode	*157*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Configure IP Address	*152*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Network Subnet Mask Configure	*153*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Network Gateway Configure	*156*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Renew DHCP	*193#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Access WEB by WAN in Route Mode	*160*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Reset Factory	*166*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Restart Device	*111#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call Function			
Call Onhold/Offhold	*#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call by IP	*47*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call Waiting Activate	*51#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call Waiting Deactivate	*50#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Blind Transfer	*87*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call Forward Unconditional Activate	*72*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call Forward Unconditional Deactivate	*73#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Call Forward Busy Activate	*90*	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Do Not Disturb Activate	*78#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Do Not Disturb Deactivate	*79#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>
Dial Voicemail	*200#	<input checked="" type="checkbox"/>	Enable <input type="button" value="▼"/>

Note: Please finish dialing the feauture code within 2s when using the 'Call holding' function.

Figure 4.8-10 Feature Code Configuration Interface

Inquire LAN port IP address	Dial*158# to obtain device WAN port IP address
Inquire WAN port IP address	Dial*159# to obtain device WAN port IP address
Inquire Phone Number	Dial*114# to obtain port account
Setting IP Mode	*150*0#, means pppmodem, *150*1#, means static IP, *150*2#, means obtain IP address by DHCP, *150*3#, means PPPoE.
Network Work Mode	*157*0#, set network work mode to routing mode; *157*1#, set network work mode to bridge mode
Configure IP Address	*152*+IP, set gateway IP address
Network subnet mask configure	*153*+subnet mask, set gateway subnet mask
Network Gateway Configure	*156*+gateway IP, set gateway
Renew DHCP	*193#, set dynamic IP again
Access Web by Wan in Rout Mode	Allow access web through WAN port: *160*1#; don't allow access web through WAN port: *160*0#
Reset Factory	*166*000000#, reset factory
Restart Device	*111#, restart device
Call onhold/offhold	When call process, dial*# into call hold. (Recovery the call through hook flash or *#)
Call by IP	Directly dial the end user IP to call
Call Waiting Activate	*51#, enable call waiting function
Call Waiting Deactivate	*50#, forbid call waiting function
Blind Transfer	If the call transfer to 801, first hook flash and then dial the * 87 * 801#
Call Forward Unconditional Activate	*72*+ phone number#, transfer the call from the phone number
Call Forward Unconditional Deactivate	*73#, forbid call forward unconditional
Call Forward Busy Activate	*90*+ forward busy number#
Call Forward Busy Deactivate	*91#, forbid call forward busy
Call Forward No Reply Activate	*92*+ forward no reply number#
Call Forward No Reply Deactivate	*93#, close this function
Do Not Disturb Activate	*78#, enable DND function
Do Not Disturb Deactivate	*79#, close DND function
Dial Voicemail	*200#, visit voice mail box

Note: * Private services are open by default

4.8.7 System Parameter

System parameters include: STUN、NTP、Provision、WEB parameter、Telnet.

1) STUN: STUN (Simple Traversal of UDP over NATs) is a network protocol. It allows users back of NAT find their own public network address, NAT type and internet end port have been bound by NAT for a local port. Two back of NAT router devices established UDP communication through this information.

STUN doesn't support TCP connection and H.323.

2) NTP: Network Time Protocol (NTP) is a computer time synchronization protocol.

System parameter configuration interface as follow:

System Parameter

STUN	<input type="checkbox"/> Enable
NTP	<input checked="" type="checkbox"/> Enable
Primary NTP Server Address	us.pool.ntp.org
Primary NTP Server Port	123
Secondary NTP Server Address	64.236.96.53
Secondary NTP Server Port	123
SYN Interval	3600 s
Time Zone	GMT-6:00 (US Central Time, Ch
Daily Reboot	<input type="checkbox"/> Enable
Reboot Time	<input type="button"/> : <input type="button"/>
WEB Parameter	
WEB Port	80
Access WEB by WAN	<input checked="" type="checkbox"/> Enable
Telnet Parameter	
Telnet Port	23

Save

Figure 4.8-11 System Configuration Interface

STUN Server Address	STUN server IP address
STUN Server Port	STUN server port
Primary NTP server address	Primary NTP server IP address, system default is us.pool.ntp.org
Primary NTP server port	Default is 123
Secondary NTP server address	Default is 18.145.0.30
Secondary NTP server port	Default is 123
SYN Interval	Every certain time synchronization gateway time, the system default every 3600 s synchronous once.
Time Zone	Time zone can be chosen. System default the United States central time, Chicago.
Reboot Time	Set a restart time for device, the device will reboot at this time.
WEB Port	Gateway web port, default is 80
Access Web by WAN	Enable or disable accessing web by WAN
Telnet Port	Telnet service port, default is 23.

4.9 Call & Routing

4.9.1 Port Group

Port group parameter include: Index, description etc. Port group configure interface as follow:

Port Group Add

Index	<input type="text" value="7"/>
Description	<input type="text"/>
Primary Display Name	<input type="text"/>
Primary SIP User ID	<input type="text"/>
Primary Authenticate ID	<input type="text"/>
Primary Authenticate Password	<input type="text"/>
Secondary Display Name	<input type="text"/>
Secondary SIP User ID	<input type="text"/>
Secondary Authenticate ID	<input type="text"/>
Secondary Authenticate Password	<input type="text"/>
Offhook Auto-Dial	<input type="text"/>
Auto-Dial Delay Time	<input type="text"/>
Port Select	<input type="text" value="Cyclic Ascending"/>
Pick Up on Group	<input type="text" value="*#"/>
Port	<input type="checkbox"/> Port 0(FXS) <input type="checkbox"/> Port 1(FXS) <input type="checkbox"/> Port 2(FXS) <input type="checkbox"/> Port 3(FXS) <input type="checkbox"/> Port 4(FXO) <input type="checkbox"/> Port 5(FXO) <input type="checkbox"/> Port 6(FXO) <input type="checkbox"/> Port 7(FXO)

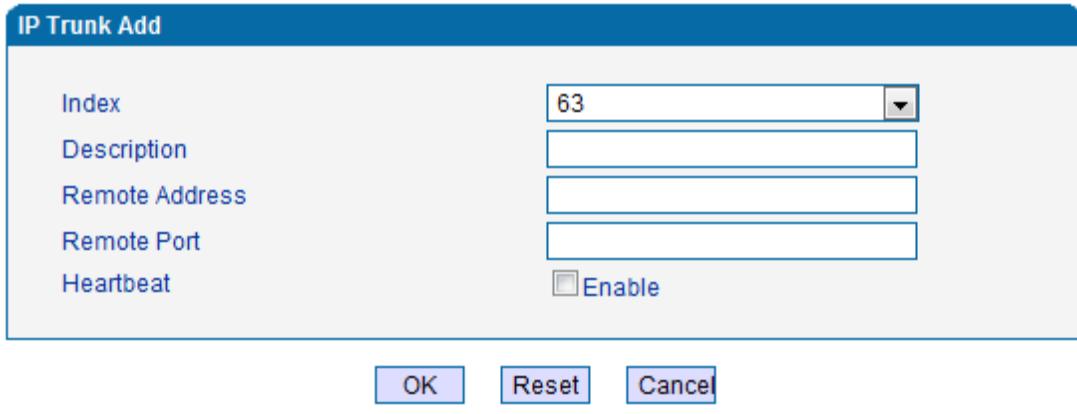
Figure 4.9-1 port group configuration interface

Index	Port group Number, It uniquely identifies a route, range from 0-7
Description	Port group description, its purpose is so you can identify the port group with a meaningful name
Primary/Secondary Display Name	Port group display, which will be used in SIP message, example: INVITE sip:bob@biloxi.com SIP/2.0 Via:SIP/2.0/UDPpc33.atlanta.com;branch=z9hG4bK776asdhs Max-Forwards: 70 To: Bob <sip:bob@biloxi.com> From: Alice <sip:alice@atlanta.com>;tag=1928301774 Here Bob and Alice is the display
Primary/Secondary SIP User ID	User account information, provided by VoIP service provider (ITSP). Usually in the form of digit similar to phone number or actually a phone number.
Primary/Secondary Authenticate ID	SIP service subscriber's Authenticate ID used for authentication. Can be identical to or different from SIP User

	ID.
Primary/Secondary Password	Authenticate
Offhook Auto-Dial	Set Auto-dial number to complete one stage dialing.
Auto-Dial delay time	Delay time of FXO port send auto-dial number.
Port Select	<ul style="list-style-type: none"> It specifies the policy for selecting port in a port group Ascending: the system always selects a port from the minimum number. The preferential selection of the port can be realized through this mode Cyclic ascending: when system selects ports' Priority, it always begin from the number next to the number selected last time, if the maximum priority number is selected last time, then the next number is the minimum priority number, and move in cycles like this Descending: when system selects ports' priority, it always begin to select from the maximum priority number Cyclic descending: when system selects ports' Priority, it always begin from the number before to the number selected last time, if the minimum priority number is selected last time, then the next number is the maximum priority number, and move in cycles like this Group ring: all ports ringing at the same time
Pick Up on Group	Press "*# +extension number" to decide which extension on the phone.
Port	Add some ports to the same group

4.9.2 IP Trunk

A peer-to-peer VoIP call occurs when two VoIP phones communicate directly over IP without IP PBXs between them. A peer-to-peer call can be initiated directly by dialing destination phone number in DAGs and also receiving incoming calls from other peer to peer gateway. IP trunk is help to DAGs establish peer-to-peer call between DAGs and other VoIP phones. IP trunk will be used in routing configuration.



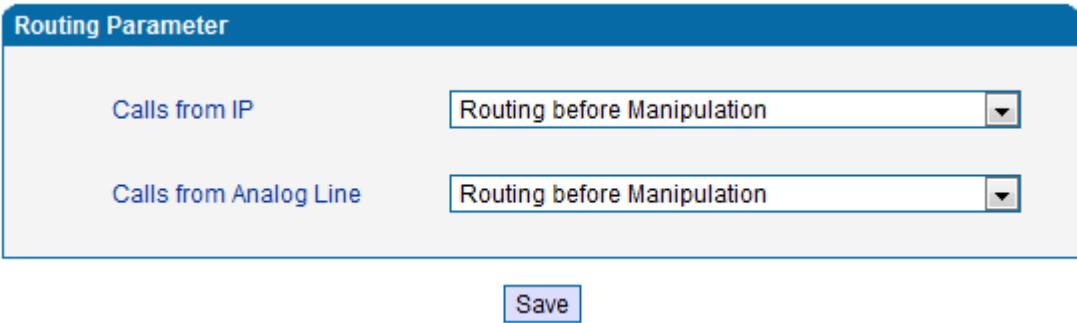
The screenshot shows the 'IP Trunk Add' configuration interface. It includes fields for 'Index' (set to 63), 'Description' (empty), 'Remote Address' (empty), 'Remote Port' (empty), and a checkbox for 'Heartbeat' (unchecked). Below the form are three buttons: 'OK', 'Reset', and 'Cancel'.

Figure 4.9-2 IP Trunk Configuration Interface

Index	IP trunk number, it is range from 0 to 63
Description	The description of IP trunk, its purpose is so you can identify the IP trunk with a meaningful name
Remote Address	Peer IP address or domain name
Remote Port	Peer SIP port
Heartbeat	Default is disable, if enable, DAG will send "OPTION" to peer device

4.9.3 Routing Configuration

Figure 4.9-3 Routing Parameter Configuration Interface



The screenshot shows the 'Routing Parameter' configuration interface. It includes two dropdown menus: 'Calls from IP' (set to 'Routing before Manipulation') and 'Calls from Analog Line' (set to 'Routing before Manipulation'). Below the dropdowns is a 'Save' button.

This option determines the following routing of call take effect before or after manipulation.

4.9.4 IP-Tel Routing

IP->Tel Routing Add

Index	31
Description	
Calls from	<input type="radio"/> IP Trunk Any <input checked="" type="radio"/> SIP Server
Caller Prefix	
Callee Prefix	
Calls to	<input type="radio"/> Port 0 <input checked="" type="radio"/> Port Group

NOTES: 'any' in 'Callee Prefix' or 'Caller Prefix' means wildcard string.

Figure 4.9-4 IP-Tel Routing Parameter

Index	Routing priority: 0-31, 0 is the highest priority.
Description	its purpose is so you can identify the IP0->Tel routing with a meaningful name
Calls from	IP Trunk/SIP Server, any means any IP
Caller Prefix	Caller number Prefix, its length normally less or equal to caller number, which helps to matching routing exactly. if caller number is 2001, the caller prefix can be 200 or 2. "any" means match any caller number like "bob1","29801"
Callee Prefix	Called number Prefix, its length normally less or equal to called number, which helps to matching routing exactly. if called number is 008675526456659, the called prefix can be 0086755 or 00. "any" means match any called number
Calls to	This call routing is routing to port or port group

4.9.5 Tel-IP/Tel Routing

Tel->IP/Tel Routing Add

Index	<input type="text" value="31"/>
Description	<input type="text"/>
Calls from	<input checked="" type="radio"/> Port <input type="text" value="0"/> <input type="radio"/> Port Group <input type="text"/>
Caller Prefix	<input type="text"/>
Callee Prefix	<input type="text"/>
Calls to	<input type="radio"/> Port <input type="text" value="0"/> <input type="radio"/> Port Group <input type="text"/> <input type="radio"/> IP Trunk <input type="text"/> <input checked="" type="radio"/> SIP Server <input type="text"/>
<input type="button" value="OK"/> <input type="button" value="Reset"/> <input type="button" value="Cancel"/>	

NOTES: 'any' in 'Callee Prefix' or 'Caller Prefix' means wildcard string.

Figure 4.9-5 Tel-IP/Tel Parameters Configuration

Index	Routing priority :0-31, 0 is the highest priority.
Description	its purpose is so you can identify the routing with a meaningful name
Calls From	Tel-IP call select port or port group
Caller Prefix	Caller number Prefix, its length normally less or equal to caller number, which helps to matching routing exactly. if caller number is 2001, the caller prefix can be 200 or 2. "any" means match any caller number like "bob1","29801"
Callee Prefix	Called number Prefix, its length normally less or equal to called number, which helps to matching routing exactly. if called number is 008675526456659, the called prefix can be 0086755 or 00., "any" means match any called number
Calls to	This call routing is routing to port, port group, IP trunk and SIP server.

4.10 Manipulation Configuration

4.10.1 IP-Tel Callee

IP->Tel Callee Add

Index	31
Description	
Calls from	<input type="radio"/> IP Trunk Any <input checked="" type="radio"/> SIP Server
Caller Prefix	
Callee Prefix	
Calls to	<input checked="" type="radio"/> Port 0 <input type="radio"/> Port Group
Stripped Digits from Left	
Stripped Digits from Right	
Prefix to Add	
Suffix to Add	
Number of Digits to Leave from Right	

OK **Reset** **Cancel**

NOTE: 'any' in 'Callee Prefix' or 'Caller Prefix' means wildcard string.

Figure 4.10-1 IP-Tel Callee number configuration

Description	IP-Tel manipulation name
Calls From	This call come from IP trunk or SIP server.
Caller Prefix	Caller number Prefix, its length normally less or equal to caller number, which helps to matching routing exactly. if caller number is 2001, the caller prefix can be 200 or 2. "any" means match any caller number like "bob1","29801"
Callee Prefix	Called number Prefix, its length normally less or equal to called number, which helps to matching routing exactly. if called number is 008675526456659, the called prefix can be 0086755 or 00., "any" means match any called number
Calls to	This call routing is routing to port, port group
Stripped Digits from Left	Remove the called number digits from the left
Stripped Digits from Right	Remove the called number digits from the right
Prefix to Add	Add a number prefix

Suffix to Add	Add a number suffix
Number of Digits to Leave from Right	Starting from the right to retain the called number digits

4.10.2 Tel-IP Caller

Tel->IP Caller Add

Index	31
Description	
Calls from	<input checked="" type="radio"/> Port 0 <input type="radio"/> Port Group
Caller Prefix	
Callee Prefix	
Calls to	<input type="radio"/> Port 0 <input type="radio"/> Port Group <input type="radio"/> IP Trunk Any <input checked="" type="radio"/> SIP Server
Stripped Digits from Left	
Stripped Digits from Right	
Prefix to Add	
Suffix to Add	
Number of Digits to Leave from Right	

NOTE: 'any' in 'Callee Prefix' or 'Caller Prefix' means wildcard string.

Figure 4. 10-2 Tel-IP Caller

Configuration parameters are the same with "IP->Tel Callee".

4.10.3 Tel-IP Callee

Tel->IP Callee Add

Index	31
Description	
Calls from	<input checked="" type="radio"/> Port 0 <input type="radio"/> Port Group
Caller Prefix	
Callee Prefix	
Calls to	<input type="radio"/> Port 0 <input type="radio"/> Port Group <input type="radio"/> IP Trunk Any <input checked="" type="radio"/> SIP Server
Stripped Digits from Left	
Stripped Digits from Right	
Prefix to Add	
Suffix to Add	
Number of Digits to Leave from Right	

OK Reset Cancel

NOTE: 'any' in 'Callee Prefix' or 'Caller Prefix' means wildcard string.

Figure 4.10-3 Tel-IP Callee
Configuration parameters are the same with "Tel->IP Caller".

4.11 Maintenance

4.11.1 Syslog Parameter

Syslog is a protocol used in (TCP/IP) network transmission of record of the standard file information.

Syslog agreement belongs to a kind of master slave agreement: Syslog sender will sent a small text information (less than 1024 bytes) to syslog the receiver. The receiver are: "syslogd", "syslog daemon" or syslog server. Syslog message can be transferred by TCP/UDP.

Syslog level:

- none Used to misarrange
- debug Not including function conditions or the question of other information
- notice importance common conditions
- warning Early warning information
- error Stop error conditions of tools or some part of the realization of the function subsystem

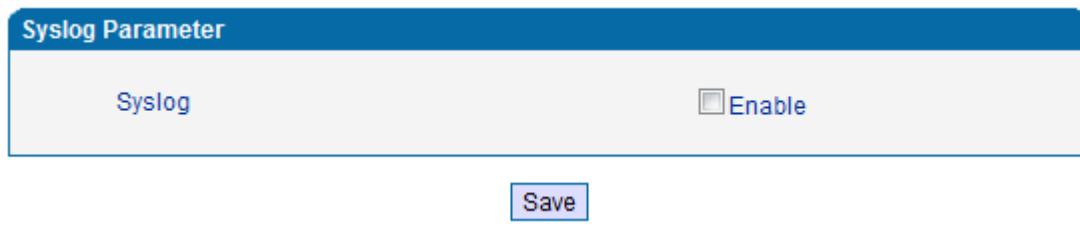


Figure 4.11-1 Syslog Parameter Configuration

Enable send CDR, and then send communication information to syslog server.

4.11.2 Firmware Upload

The process of firmware upload:

- 1) Click "Firmware Upload"
- 2) Browse files and choose the loading program (Name the file extension. Idf)
- 3) Click "Upload", the upload process will last about 60s and device can automatically restart after uploading. (The firmware update process don't shut off the power)



Notes: 1. The upload process will last about 60s.
2. The device will restart automatically after upload.
3. Do not shut down when the device is uploading.

Figure 4.11-2 Firmware upload Configuration

4.11.3 Data Backup

The process data backup:

- 1) Click "Data Backup"
- 2) Click "Backup" to backup data to PC.

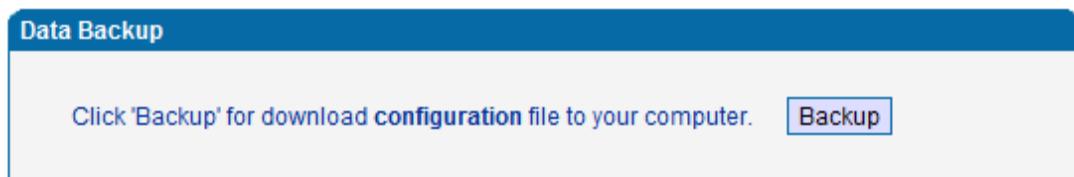


Figure 4.11-3 Data Backup Interface

4.11.4 Data Restore

The processes of data restore:

- 1) Click "Data Restore"
- 2) Browse file, select data file.
- 3) Click "Restore" and then import successfully, the device will restart automatically.



Figure 4.11-4 Data Restore Interface

4.11.5 Ping Test

Send test data packets to IP, check each other whether have response and statistical response time. It is ping. Used to test internet and analyzed network fault.

Application format: Ping IP address. It is used to check the network connectivity or network connection speed command.

Ping instructions:

- 1) Click "ping test"

- 2) Fill IP address or domain connected, click start.
- 3) Received a message indicates that network connection normal, or network connected to a fault.

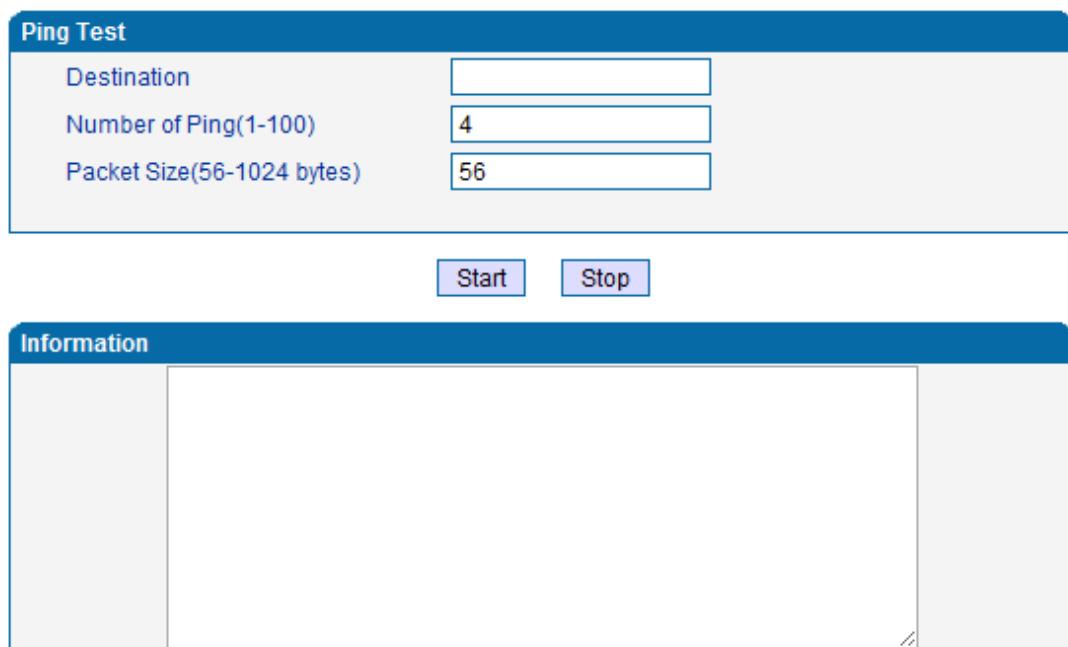


Figure 4.11-5 Ping Parameter Interface

4.11.6 Tracert Test

Tracert is trace router and used to tracking routing.

Tracert sends a sequence of Internet Control Message Protocol (ICMP) echo request packets addressed to a destination host. Determining the intermediate routers traversed involves adjusting the time-to-live (TTL), aka hop limit, Internet Protocol parameter. Frequently starting with a value like 128 (Windows) or 64 (Linux), routers decrement this and discard a packet when the TTL value has reached zero, returning the ICMP error message ICMP Time Exceeded. Tracert works by increasing the TTL value of each successive set of packets sent. The first set of packets sent have a hop limit value of 1, expecting that they are not forwarded by the first router. The next set have a hop limit value of 2, so that the second router will send the error reply. This continues until the destination host receives the packets and returns an ICMP Echo Reply message.

Trace route uses the returned ICMP messages to produce a list of hops (which usually consists

of routers and layer 3 switches) that the packets have traversed. The timestamp values returned for each router along the path are the delay (aka latency) values, typically measured in milliseconds for each packet.

Tracert introduce:

- 1) Click tracert test.
- 2) Fill IP address or domain connected, click start.

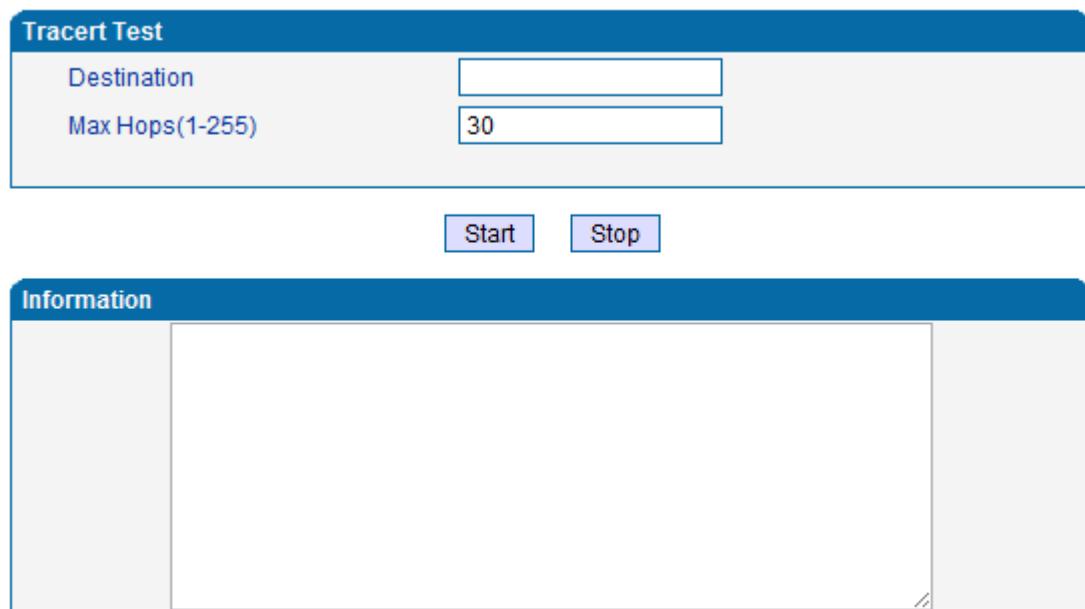


Figure 4.11-6 Tracert Test Interface

4.11.7 Password Modification

Includes WEB username and password, Telnet username and password modify.

Note: Default web and telnet username and password is: admin, admin.

Password Modification

Web Config	
Old Web Username	<input type="text" value="admin"/>
Old Web Password	<input type="text"/>
New Web Username	<input type="text"/>
New Web Password	<input type="text"/>
Confirm Web Password	<input type="text"/>
Telnet Config	
Old Telnet Username	<input type="text" value="admin"/>
Old Telnet Password	<input type="text"/>
New Telnet Username	<input type="text"/>
New Telnet Password	<input type="text"/>
Confirm Telnet Password	<input type="text"/>

Save

Figure 4.11-7 Password Modification Interface

4.11.8 Factory Reset

Click “Apply” to restore the factory settings.

Factory Reset

Click the button below to reset to factory default settings.

Apply

Figure 4. 11-8 Factory Reset Interface

4.11.9 Device Restart

Click the “Save” button in the Configuration page to save the changes to the equipment configuration. The following screen confirms that the changes are saved. If the changes need restart, reboot or power cycle the equipment to make the changes take effect.

Restart

Click the button below to restart the device.

Restart

Figure 4.11-9 Device Restart

5. Glossary

- DNS: Domain Name System
- SIP: Session Initiation Protocol
- TCP: Transmission Control Protocol
- UDP: User Datagram Protocol
- RTP: Real Time Protocol
- PPPOE: point-to-point protocol over Ethernet
- VLAN: Virtual Local Area Network
- ARP: Address Resolution Protocol
- CID: Caller Identity
- DND: Do NOT Disturb
- DTMF: Dual Tone Multi Frequency
- NTP: Network Time Protocol
- DMZ: Demilitarized Zone
- STUN: Simple Traversal of UDP over NAT
- PSTN: Public Switched Telephone Network